APPENDIX A
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TELOGY™ NETWORKS

Voice Over Packet

White Paper

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Introduction

Organizations around the world want to reduce rising communications costs. The consolidation of separate voice and data networks offers an opportunity for significant savings. Accordingly, the challenge of integrating voice and data networks is becoming a rising priority for many network managers. Organizations are pursuing solutions which will enable them to take advantage of excess capacity on broadband networks for voice and data transmission, as well as utilize the Internet and company Intranets as alternatives to costlier mediums.

A Voice Over Packet application meets the challenges of combining legacy voice networks and packet networks by allowing both voice and signaling information to be transported over the packet network. This paper references a general class of packet networks since the modular software objects allow networks such as ATM, Frame Relay, and Internet/Intranet (IP) to transport voice. An overview of a software architecture utilizing Embedded Communication Objects™ (ECOs™) that support Voice Over Packet applications is presented.

ECOs are real-time software and hardware modules that can be dynamically configured to provide flexibility and scalability in communication systems. ECOs have well defined Application Programming Interfaces (APIs), support multiple channel operation through the use of an instance structure, and use a dynamic binding mechanism that allows flexibility in system configuration. Customers can gain a considerable advantage in time to market by using ECOs in building their communication systems.

As shown in Figure 1, the legacy telephony terminals that are addressed range from standard two wire Plain Old Telephone Service (POTS) and Fax Terminals to digital and analog PBX interfaces. Packet networks supported are ATM, Frame Relay, and Internet.

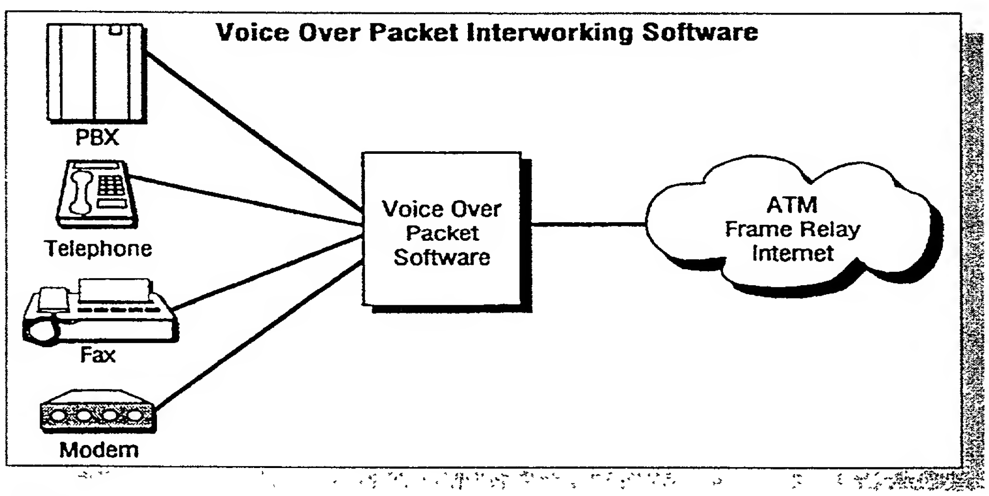

Figure 1

Applications

A wide variety of applications are enabled by the transmission of Voice Over Packet networks. This paper will explore three examples of these applications.

The first application, shown in Figure 2, is a network configuration of an organization with many branch offices (e.g. a bank) that wants to reduce costs and combine traffic to provide voice and data access to the main office. This is accomplished by using a packet network to provide standard data transmission while at the same time enhancing it to carry voice traffic along with the data. Typically, this network configuration will benefit if the voice traffic is compressed due to the low bandwidth available for this access application. Voice Over Packet provides the Interworking Function (IWF), which is the physical implementation of the hardware and software that allows the transmission of combined voice and data over the packet network. The interfaces the IWF must support in this case are analog interfaces which directly connect to telephones or Key systems.

The IWF must emulate the functions of both a PBX for the telephony terminals at the branches, as well as the functions of the telephony terminals for the PBX at the home office. The IWF accomplishes this by implementing signaling software that performs these functions.

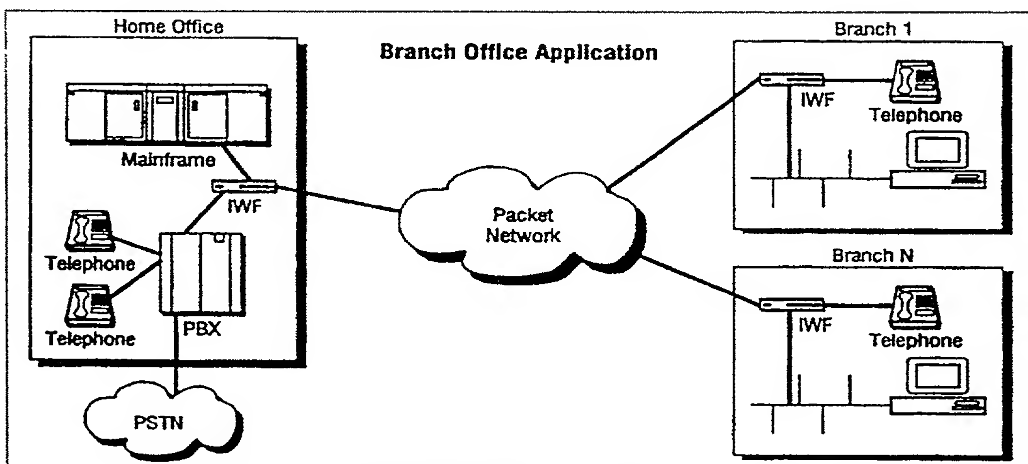

Figure 2

A second application of Voice Over Packet, shown in Figure 3, is a trunking application. In this scenario, an organization wants to send voice traffic between two locations over the packet network and replace the Tie Trunks used to connect the PBXs at the locations. This application usually requires the Interworking Function to support a higher capacity digital channel than the branch application, such as a T1/E1 interface of 1.544 or 2.048 Mbps. The Interworking Function emulates the signaling functions of a PBX, resulting in significant savings to companies' communications costs.

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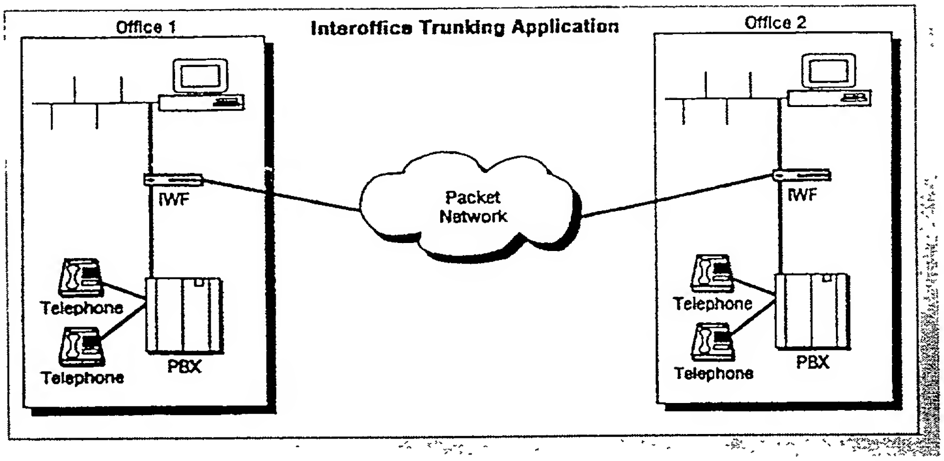

Figure 3

A third application of Voice Over Packet software is interworking with Cellular Networks, as shown in Figure 4. The voice data in a digital cellular network is already compressed and packetized for transmission over the air by the cellular phone. Packet networks can then transmit the compressed cellular voice packet, saving a tremendous amount of bandwidth. The IWF provides the transcoding function required to convert the cellular voice data to the format required by the Public Switched Telephone Network (PSTN).

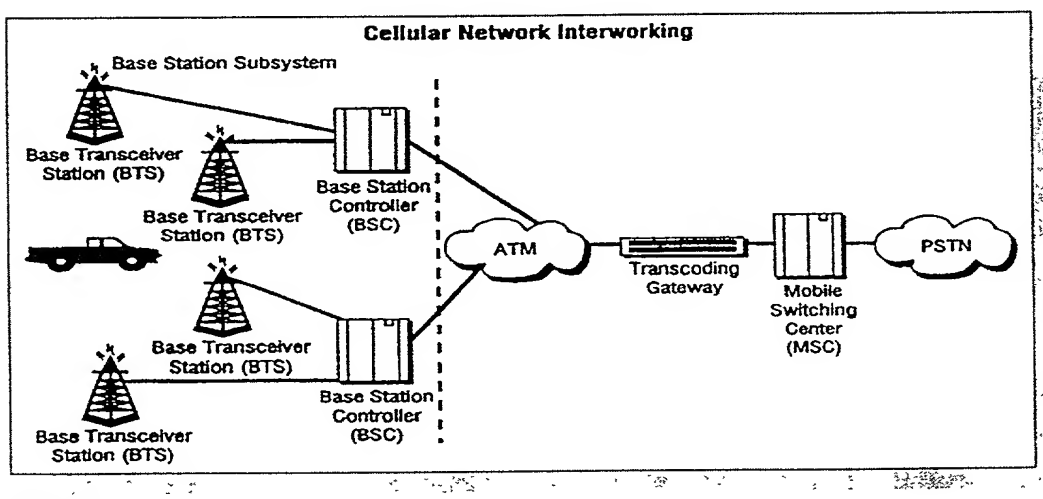

Figure 4

Quality of Service

The advantages of reduced cost and bandwidth savings of carrying Voice Over Packet networks are associated with some quality of service issues unique to packet networks. These issues are explored below.

Delay

Delay causes two problems — echo and talker overlap.

Echo is caused by the signal reflections of the speaker's voice from the far end telephone equipment back into the speaker's ear. Echo becomes a significant problem when the round trip delay becomes greater than 50 milliseconds. Since echo is perceived as a significant quality problem, Voice Over Packet systems must address the need for echo control and implement some means of echo cancellation.

Talker overlap (or the problem of one talker stepping on the other talker's speech) becomes significant if the one-way delay becomes greater than 250 msec. The end-to-end delay budget is therefore the major constraint and driving requirement for reducing delay through a packet network.

Following are sources of delay in an end to end Voice Over Packet call:

1. **Accumulation Delay** (sometimes called algorithmic delay): This delay is caused by the need to collect a frame of voice samples to be processed by the voice coder. It is related to the type of voice coder used and varies from a single sample time (.125 microseconds) to many milliseconds. A representative list of standard voice coders and their frame times follows:
   - G.726 ADPCM (16, 24, 32, 40 Kbps) - .125 microseconds
   - G.728 - LD-CELP(16 Kbps) - 2.5 milliseconds
   - G.729 - CS- ACELP (8 Kbps) - 10 milliseconds
   - G.723.1 - Multi Rate Coder (5.3, 6.3 Kbps) - 30 milliseconds
2. **Processing Delay:** This delay is caused by the actual process of encoding and collecting the encoded samples into a packet for transmission over the packet network. The encoding delay is a function of both the processor execution time and the type of algorithm used. Often, multiple voice coder frames will be collected in a single packet to reduce the packet network overhead. For example, three frames of G.729 codewords, equaling 30 milliseconds of speech, may be collected and packed into a single packet.
3. **Network Delay**: This delay is caused by the physical medium and protocols used to transmit the voice data, and by the buffers used to remove packet jitter on the receive side. Network delay is a function of the capacity of the links in the network and the processing that occurs as the packets transit the network. The jitter buffers add delay which is used to remove the packet delay variation that each packet is subjected to as it transits the packet network. This delay can be a significant part of the overall delay since packet delay variations can be as high as 70-100 msec in some Frame Relay networks and IP networks.

Jitter

The delay problem is compounded by the need to remove jitter, a variable inter-packet timing caused by the network a packet traverses. Removing jitter requires collecting packets and holding them long enough to allow the slowest packets to arrive in time to be played in the correct sequence. This causes additional delay.

The two conflicting goals of minimizing delay and removing jitter have engendered various schemes to adapt the jitter buffer size to match the time varying requirements of network jitter removal. This adaptation has the explicit goal of minimizing the size and delay of the jitter buffer, while at the same time preventing buffer underflow caused by jitter.

Two approaches to adapting the jitter buffer size are detailed below. The approach selected will depend on the type of network the packets are traversing.

1. The first approach is to measure the variation of packet level in the jitter buffer over a period of time, and incrementally adapt the buffer size to match the calculated jitter. This approach works best with networks that provide a consistent jitter performance over time, such as ATM networks.
2. The second approach is to count the number of packets that arrive late and create a ratio of these packets to the number of packets that are successfully processed. This ratio is then used to adjust the jitter buffer to target a predetermined allowable late packet ratio. This approach works best with the networks with highly variable packet inter-arrival intervals, such as IP networks.

In addition to the techniques described above, the network must be configured and managed to provide minimal delay and jitter, enabling a consistent quality of service.

Lost Packet Compensation

Lost packets can be an even more severe problem, depending on the type of packet network that is being used. Because IP networks do not guarantee service, they will usually exhibit a much higher incidence of lost voice packets than ATM networks. In current IP networks, all voice frames are treated like data. Under peak loads and congestion, voice frames will be dropped equally with data frames. The data frames, however, are not time sensitive and dropped packets can be appropriately corrected through the process of retransmission. Lost voice packets, however, cannot be dealt with in this manner.

Some schemes used by Voice Over Packet software to address the problem of lost frames are:

1. Interpolate for lost speech packets by replaying the last packet received during the interval when the lost packet was supposed to be played out. This scheme is a simple method that fills the time between non-contiguous speech frames. It works well when the incidence of lost frames is infrequent. It does not work very well if there are a number of lost packets in a row or a burst of lost packets.

2. Send redundant information at the expense of bandwidth utilization. The basic approach replicates and sends the nth packet of voice information along with the (n+1)th packet. This method has the advantage of being able to exactly correct for the lost packet. However, this approach uses more bandwidth and also creates greater delay.

3. A hybrid approach uses a much lower bandwidth voice coder to provide redundant information carried along in the (n+1)th packet. This reduces the problem of the extra bandwidth required, but fails to solve the problem of delay.

Echo Compensation

Echo in a telephone network is caused by signal reflections generated by the hybrid circuit that converts between a 4-wire circuit (a separate transmit and receive pair) and a 2-wire circuit (a single transmit and receive pair). These reflections of the speaker's voice are heard in the speaker's ear. Echo is present even in a conventional circuit switched telephone network. However, it is acceptable because the round trip delays through the network are smaller than 50 msec. and the echo is masked by the normal side tone every telephone generates.

Echo becomes a problem in Voice Over Packet networks because the round trip delay through the network is almost always greater than 50 msec. Thus, echo cancellation techniques are always used. ITU standard G.165 defines performance requirements that are currently required for echo cancellers. The ITU is defining much more stringent performance requirements in the G.IEC specification.

Echo is generated toward the packet network from the telephone network. The echo canceller compares the voice data received from the packet network with voice data being transmitted to the packet network. The echo from the telephone network hybrid is removed by a digital filter on the transmit path into the packet network.

Software Architecture

Two major types of information must be handled in order to interface telephony equipment to a packet network — voice and signaling information.

As shown in Figure 5, Voice Over Packet software interfaces to both streams of information from the telephony network and converts them to a single stream of packets transmitted to the packet network.

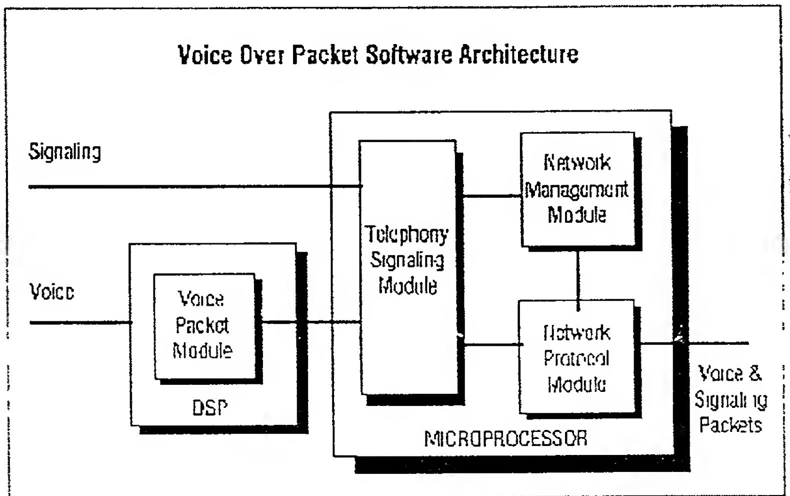

Figure 5

The software functions are divided into four general areas:

1. **Voice Packet Module:** This software, typically run on a DSP, prepares voice samples for transmission over the packet network. Its components perform tone detection and generation, echo cancellation, voice compression, voice activity detection, jitter removal, resampling, and voice packetization.
2. **Telephony Signaling Gateway Module:** This software interacts with the telephony equipment, translating signaling into state changes used by the Network Protocol Module (described below) to set up connections. These state changes are on-hook, off-hook, trunk seizure, etc. This software supports E&M (wink, delay and immediate), Loop or Ground Start FXS and FXO, ISDN BRI/PRI and QSIG.
3. **Network Protocol Module:** This module processes signaling information and converts it from the telephony signaling protocols to the specific packet signaling protocol used to set up connections over the packet network (e.g. Q.933 and Voice over FR signaling). It also adds protocol headers to both voice and signaling packets before transmission into the packet network.
4. **Network Management Module:** This module provides the voice management interface to configure and maintain the other modules of the Voice

Over Packet system. All management information is defined in ASN.1 and complies with SNMP V1 syntax. A proprietary Voice Packet MIB is supported until standards evolve in the Forums.

The software is partitioned to provide a well defined interface to the DSP software usable for multiple voice packet protocols and applications. The DSP processes voice data and passes voice packets to the microprocessor with generic voice headers. The microprocessor is responsible for moving voice packets and adapting the generic voice headers to the specific Voice Packet Protocol that is called for by the application, such as Real Time Protocol (RTP), Voice over Frame Relay (VOFR), and Voice Telephony over ATM (VTOA). The microprocessor also processes signaling information and converts it from supported telephony signaling protocols to the packet network signaling protocol (e.g. H.323 (IP), Frame Relay, or ATM signaling).

This partitioning provides a clean interface between the generic voice processing functions — such as compression, echo cancellation, and voice activity detection — and the application specific signaling and voice protocol processing.

Voice Packet Module

This section describes the functions performed by the software in the Voice Packet Module, which is primarily responsible for processing the voice data. This function is usually performed in a Digital Signal Processor (DSP).

The Voice Packet Module consists of the following software:

- **PCM Interface:** Receives PCM samples from the digital interface and forwards them to appropriate DSP software modules for processing. Forwards processed PCM samples received from various DSP software modules to the digital interface. Performs continuous phase resampling of output samples to the digital interface to avoid sample slips.
- **Tone Generator:** Generates DTMF tones and call progress tones under command of the Host (e.g. telephone, fax, modem, PBX or telephone switch). Configurable for support of U.S. and international tones.
- **Echo Canceller:** Performs G.165 compliant echo cancellation on sampled, full-duplex voice port signals. Programmable range of tail lengths.
- **Voice Activation Detector/Idle Noise Measurement:** Monitors the received signal for voice activity. When no activity is detected for the configured period of time, the software informs the Packet Voice Protocol. This prevents the encoder output from being transported across the network when there is silence, resulting in additional bandwidth savings. This software also measures the Idle Noise characteristics of the telephony interface. It reports this information to the Packet Voice Protocol in order to relay this information to the remote end for noise generation when no voice is present.

- **Tone Detector:** Detects the reception of DTMF tones and performs voice/fax discrimination. Detected tones are reported to the Host so that the appropriate speech or fax functions are activated.
- **Voice Codec Software:** Compresses the voice data for transmission over the packet data. Capable of numerous compression ratios through the modular architecture. A compression ratio of 8:1 is achievable with the G.729 voice codec (thus, the normal 64 Kbps PCM signal is transmitted using only 8 Kbps).
- **Fax Software:** Performs a Fax Relay function by demodulating PCM data, extracting the relevant information, and packing the fax line scan data into frames for transmission over the packet network. Significant bandwidth savings can be achieved by this process.
- **Adaptive Playout Unit:** Buffers voice packets received from the packet network and sends them to the Voice Codec for playout. The following features are supported:
  - A FIFO buffer that stores voice codewords before playout removes timing jitter from the incoming packet sequence.
  - A continuous-phase resampler that removes timing frequency offset without causing packet slips or loss of data for voice or voiceband modem signals.
  - A timing jitter measurement which allows adaptive control of FIFO delay.

  The voice packetization protocols use a sequence number field in the transmit packet stream to maintain temporal integrity of voice during playout. Using this approach, the transmitter inserts the contents of a free-running, modulo-16 packet counter into each transmitted packet, allowing the receiver to detect lost packets and to properly reproduce silence intervals during playout.
- **Packet Voice Protocol:** Encapsulates compressed voice and fax data for end-to-end transmission over a backbone network between two ports.
- **Message Processing Unit:** Coordinates the exchange of Monitor and Control information between the DSP and Host via a mailbox mechanism. Information exchanged includes software downline load, configuration data, and status reporting.
- **Real-Time Portability Environment:** Provides the operating environment for the software residing on the DSP. Provides synchronization functions, task management, memory management, and timer management.

Figure 6 diagrams the architecture of the DSP software. The DSP software processes PCM samples from the telephony interface and converts them to a digital format suitable for transmission through a packet network.

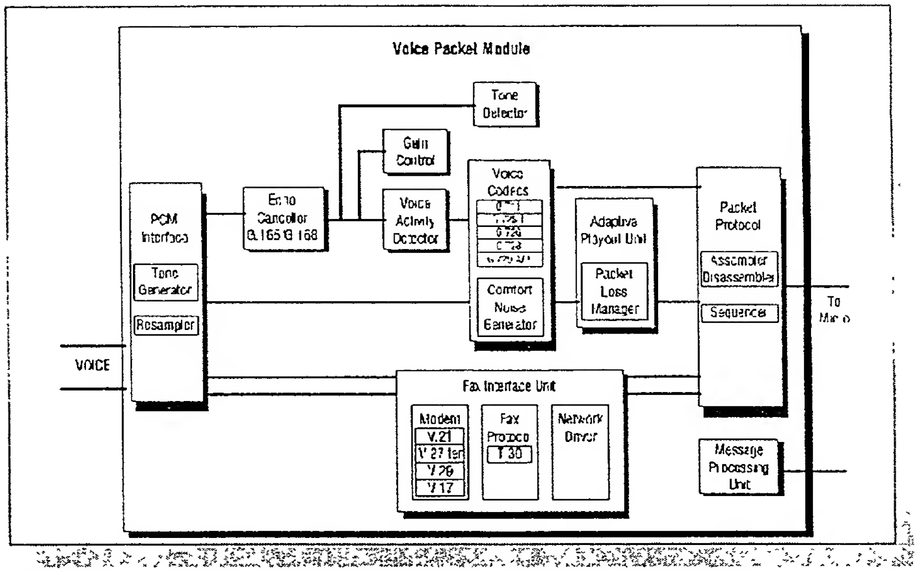

Figure 6

Signaling, Protocol, and Management Modules

The Voice Over Packet software performs telephony signaling to detect the presence of a new call and to collect address (dial digit) information which is used by the system to route a call to a destination port. It supports a wide variety of telephony signaling protocols and can be adaptable to many environments. The software and configuration data for the voice card can be downloaded from a network management system to allow customization, easy installation, and remote upgrades.

The software interacts with the DSP for tone detection and generation as well as mode of operation control based on the line supervision, and interacts with the telephony interface for signaling functions. The software receives configuration data from the network management agent and utilizes operating system services.

Figure 7 diagrams the architecture of the signaling software. The software consists of the following components:

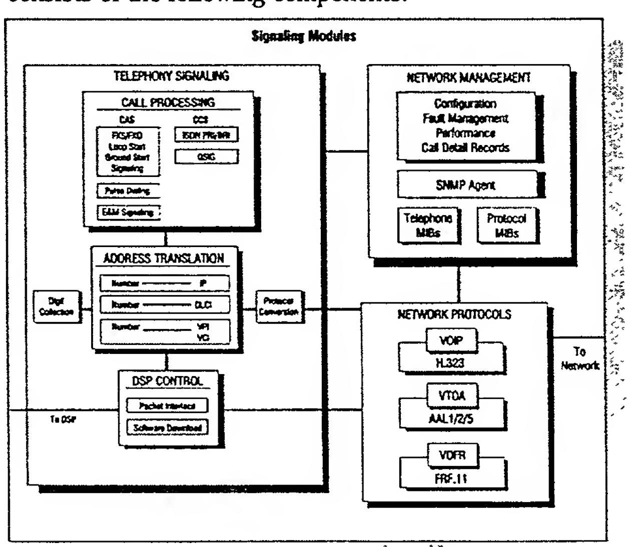

Figure 7

Telephony Signaling Gateway Module

- **Telephony Interface Unit Software:** Periodically monitors the signaling interfaces of the module and provides basic debouncing and rotary digit collection for the interface.
- **Signaling Protocol Unit:** Contains the state machines implementing the various telephony signaling protocols such as E&M.
- **Network Control Unit:** Maps telephony signaling information into a format compatible with the packet voice session establishment signaling protocol.
- **Address Translation Unit:** Maps the E.164 dial address to an address that can be used by the packet network (e.g. an IP address or a DLCI for a Frame Relay Network).
- **DSP Interface Driver:** Relays control information between the Host microprocessor and DSPs.
- **DSP Downline Loader:** Responsible for downline load of the DSPs at start-up, configuration update, or mode changes (e.g. switching from voice mode to fax mode when fax tones are detected).

Network Protocol Module

- **IP Signaling Stack:** H.323 call control and transport software including H.225, H.245 and RTP/RTCP transport protocol, TCP, IP, UDP protocols.
- **ATM Signaling Protocol Stack:** ATM Forum VTOA Voice Encapsulation Protocol. ATM Forum compliant User-Network Interface (UNI) signaling protocol stack for establishing, maintaining, and clearing point-to-point and point-to-multipoint switched virtual connections (SVCs).
- **Frame Relay Protocol Stack:** Frame Relay Forum VOFR Voice Encapsulation Protocol, PVC and SVC Support, Local Management Interface (LMI), Congestion Management and Traffic Monitoring, CIR Enforcement and Congestion.

Network Management Module

The Network Management software consists of three major services addressed in the MIB:

- Physical interface to the telephone endpoint.
- Voice channel service for:
  - processing signaling on a voice channel
  - converting between PCM samples and compressed voice packets
- Call control service for parsing call control information and establishing calls between telephony endpoints.

The Voice Over Packet software is configured and maintained through the use of a proprietary Voice Service MIB.

Summary

A Voice Over Packet software architecture using Embedded Communication Objects (ECOs) has been described for the interworking of legacy telephony systems and packet networks. Some of the key features enabling this application to function successfully are:

- an approach that minimizes the effects of delay on voice quality.
- an adaptive playout to minimize the effect of jitter.
- features that address lost packet compensation and echo cancellation.
- a flexible DSP system architecture that manages multiple channels per single DSP.

Carrying Voice Over Packet networks provides the most bandwidth efficient method of integrating these divergent technologies. While the challenges to this integration are substantial, the potential savings make the investment in a quality implementation compelling.

ABOUT TELOGY NETWORKS

Telogy Networks is the leading provider of embedded communications software to global equipment manufacturers. Telogy's Golden Gateway™ Voice over IP software enables manufacturers to develop connected products that can send real-time voice, fax, and data over multiple packet networks (such as Internet/Intranet, Frame Relay, and ATM). As one of the few embedded software companies with both DSP and microprocessor expertise, Telogy Networks offers its customers truly comprehensive product solutions.

TELOGY NETWORKS *The Leading Provider of Embedded Communications Software*

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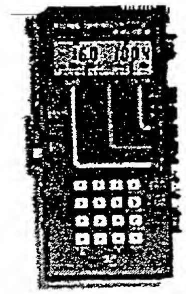
AM48 Hand-Held TIMS



What Is It?

The Ameritec Model AM48 is a compact, user-friendly, yet powerful hand-held analog transmission test set. About the size of a personal organizer and weighing less than 2 lbs., it can easily be transported in a briefcase or slide into a tool kit. It is powered by four AA batteries, an accessory 110 VAC adapter/recharger or four AA rechargeable NiCad batteries.

The unit is designed for field testing and measurement of voice and data for 2 or 4 wire telecommunications transmission circuits. This unit is a full function signal generator able to generate all of the test tones and waveforms needed to perform a variety of standard measurements. It is a telephone set with built-in dial, talk and listen capability with a selection of 600 ohm or 900 ohm termination impedance. This unit fully complies with Bell Standard 41009.

What Does It Do?

The Ameritec Model AM48 can do a variety of measurements. With its built-in dial capability on dial-up networks it can access a distant responder or a second AM48 for centralized 2-wire testing. You can use a single unit with an Ameritec responder for centralized loop back testing. Or, use two sets for a complete end-to-end test of 4-wire data lines.

In addition, the AM-8 now combined with the new AM440 Remote Test Partner™™ is able to further enhance your testing capabilities. A single technician can test a circuit with an AM48, AM440 and a multi-meter. The AM48 can transmit to the AM440 and measure its signal and can take noise measurements. The combination of these two measuring instruments allows for DMM testing, including resistance testing and the measurements of both voltage and capacitance.

How Does It Work?

The user utilizes the color-coded controls and menu selection to

select the desired measure mode and send mode. Measurements are captured and are automatically displayed. Immediate display indicates if the measurement is out of range over or under. All controls are edge-mounted rocker switches or slide switches color coded with descriptive labeling on the front of the unit for all switch functions. The keypad which is normally used for dialing, has a unique secondary function which allows setting of all control parameters associated with the more complex tests. The keypad is also able to set signal generator levels and frequencies.

What Can It Test?

A single unit with the capability to do the measurements of three separate instruments. Able to measure transmission circuits over the extended voice band 200 Hz to 20 kHz as follows:

- dBm level or dB loss
- Frequency
- Frequency Response (Attenuation Distortion)
- Noise
- Noise with Tone
- Signal/Noise Ratio
- P/AR
- 3 Level Impulse Noise
- Phase and Gain Jitter
- Hits and Dropouts

In addition, it is equipped with a user-programmable, non-volatile memory contained within the unit. The ability to store up to ten operator-defined send/measure test configurations, as well as up to 10 operator-defined test tone frequencies. Easily obtained stored data retrieved by a single keystroke from the user.

Related Products:

AM44
AM440 Remote Test Partner™
AM47 Printers
Model 401

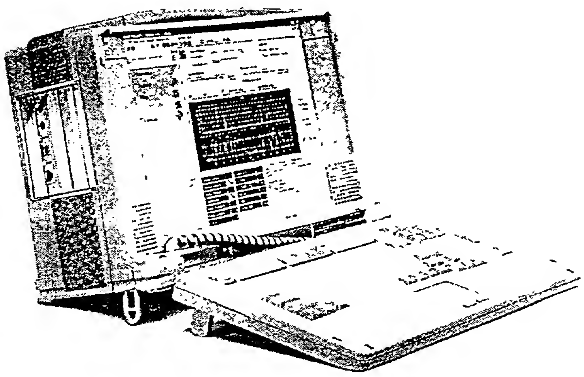
Model 402
Model 412
AM5 Classic-200
AM5XT-200

Questions about the **AM48 Hand-Held TIMS**?

Ask ? Zeke

Agilent Technologies
Telegra R - VQT J1981A

Product Overview



The Comprehensive, Objective Voice Quality Tester

The Agilent Telegra VQT is an objective, end-to-end voice quality test system providing detailed test and analysis capabilities for voice quality on modern telephony networks, including VoIP, VoATM, and PSTN. Analog, T1, and E1 interfaces connect to the access points of the network to determine the key end-user quality parameters of any voice network: clarity, delay, echo, and others. The portable system is designed for engineers who develop, deploy and operate next generation voice network devices and services. The self-guiding user interface allows expert and novice users to easily execute tests and determine the success or failure of the test. Insightful reports and graphs will aid the user to quickly and easily:

- Identify faulty system components in voice networks and network devices
- Improve network and system performance
- Shorten development, deployment and troubleshooting

The Telegra VQT is a portable tester based on the Telegra R platform. To be used as a benchtop tester in the lab or portable tester in the field it is well suited when you:

- Develop voice systems such as voice gateways or PBXs
- Validate voice quality in QA and system integration
- Integrate voice gateways into IP, ATM or Frame Relay networks
- Install voice systems in public or enterprise networks or customer sites
- Troubleshoot voice networks in the field
- Operate and maintain voice networks

Agilent Technologies
Innovating the HP Way

Introduce New Products and Network Services Sooner

The Telegra VQT helps you get products and services to market earlier with higher confidence. Locate subtle problems early in development or deployment when they are much less expensive to fix. Telegra VQT's comprehensive test capabilities enhance product and service quality and reliability.

Shorten development, deployment, and repair times

- Fine tune your network equipment under real network conditions before field deployment - Measure the network behavior in a real network and simulate this behavior through the Telegra VQT in your lab
- Simulate network changes in the lab before deployment - Use previously captured network behavior and change network parameters such as delay e.g. to simulate additional nodes or longer routes
- Automate voice quality tests - Use provided test scripts, modify existing ones or develop your own
- Let your voice expert technicians/engineers troubleshoot and validate deployments throughout the country remotely - Remotely log into the VQT and access all functionality as if you were sitting right in front of the Telegra VQT

Objectively Compare and Improve Network and Systems Performance

For voice services of next generation networks such as VoIP to be accepted by consumers and corporate users, they have to provide comparable voice quality as the traditional phone networks. Telegra VQT provides you with the ability to objectively compare voice quality and to identify the influencing factors.

- Objectively measure voice quality from an end-user perspective - recreate the user experience by measuring end-to-end quality, connecting as close to your end-user as possible using analog FXO, analog E&M, T1, or E1 interfaces
- Fully load a dual-port T1 or E1 interface and measure voice quality and performance of a network or device under various traffic conditions
- Identify network components or behavior requiring improvements - after determining poor voice quality analyze in detail the clarity, delay, echo, silence suppression, comfort noise generation and DTMF tone transmission to identify what needs to be improved
- Determine the effect of network or system changes (e.g. traffic load or design changes) by easily comparing measurement results before and after the change

Key Functionality

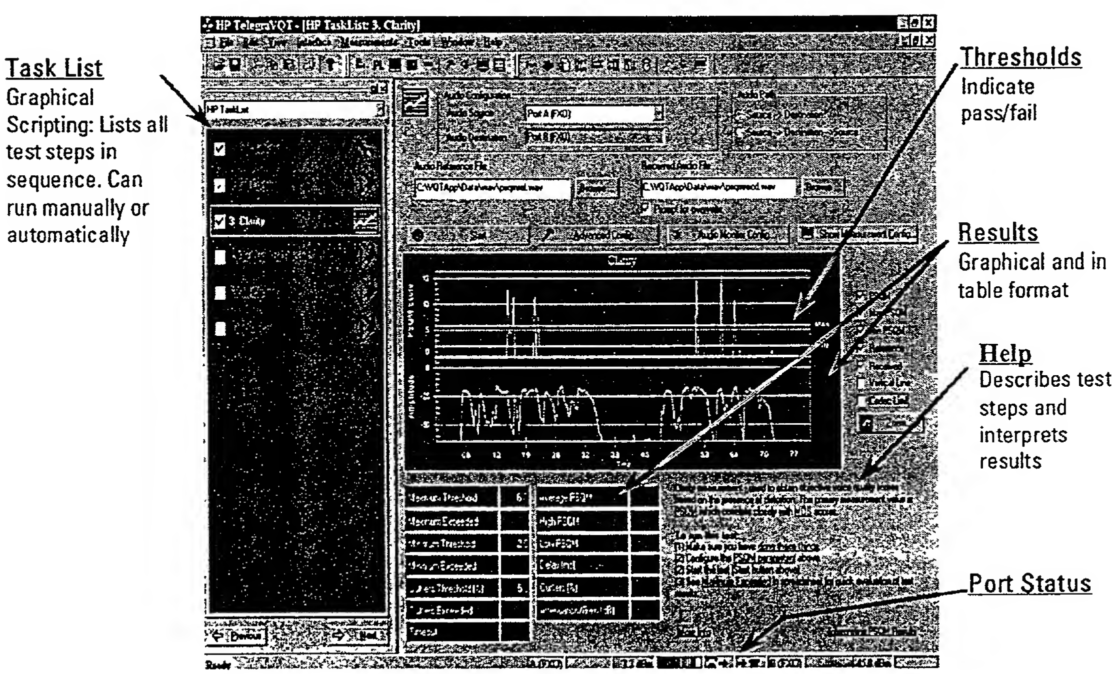
- Clarity
  - Speech quality measurement using the Perceptual Analysis Measurement System (PAMS). PAMS is an innovative technique based on a perceptual model of human hearing. The impacts of a wide range of network-induced distortions, including voice coding and packet loss, are measured and reflected in scores that correlate to Mean Opion Scores on a scale of 1-5.
  - PAMS graphs the error surface showing signal loss and additive distortion, over the time and frequency domains of the test signal.
  - Speech Quality measurement via PSQM+ - based on ITU-T P.861 but improved for network effects such as severe distortions and time clipping which can be generated through packet loss.
  - Presents PSQM scores graphically over time for the entire measurement period, correlated with the test and reference signals."
  - Generates and analyzes PSQM with real speech, including 144 different voice samples in 8 different languages: Japanese, English-North America, English/Britain, French, German, Spanish, Chinese/Beijing (Mandarin) and Chinese/Canton (Cantonese).
- Delay Measurement
  - High reliability via VF signal cross-correlation.
  - One millisecond resolution.
  - Single delay measurement and trend analysis.
- Echo Measurements
  - Detect echo in a voice transmission and measure its impact on clarity.
  - Measure performance of echo cancelers under conditions of Doubletalk.

- Voice Activation Detection Analysis - determines the effectiveness of the voice activity detector (VAD) by measuring
  - Silence suppression - front end clipping (FEC) and holdover time (HOT)
  - Comfort noise generation match with background noise
- DTMF Tone Analysis -analysis of DTMF tone degradation through a network by graphing the distortion parameters including attenuation, twist, and frequency shift
- Capture and analyze network characteristics - captures and graphs the time response of a tail-end circuit or any other linear network
- Network Simulation - simulate networks in your lab using previously measured network characteristic
- Provides analog FXO, analog E&M, T1, E1, and ISDN PRI interfaces to test at different access points and across different types of networks
- Generate calls and load traffic on multiple T1/E1 channels simultaneously, to test voice quality under various traffic conditions
- Automated task lists enable unattended testing
  - Fast and efficient testing
  - Great for setting up tests by experts, to be ran off-site or by novice users
- Built for novice and expert users –
  - A self-explanatory user interface and pre-designed scripts allow you to easily execute every test and identify problems quickly
  - Experts have access to all test parameters and settings, locally or remotely. Test scripts can be developed or existing ones changed. (user can store hundreds of tests on a single VQT)

You Do Not Need To Be A Voice Quality Expert To Operate This System!

Today, more developers and field engineers than ever are faced with the challenge of analyzing voice quality. For the Telegra VQT, ease of use was one of the primary design goals A self-explanatory user interface and pre-designed scripts allow the user to easily execute every test and identify problems quickly. Default parameters are set for every test and an integrated help window explains test steps and gives guidance on how to interpret results. On the other hand, experts have access to all test parameters and result details than enable them to extract even more information.

Flat User Interface - configuration and measurement information on one screen
Same concept and layout for all measurements



Telegra VQT provides a consistent task oriented user interface across all measurements. A taskbar allows users to execute all measurements in the correct order and to switch easily between measurements. All measurements provide configuration and result information on the same screen. Results are represented graphical and as text at the same time. User definable thresholds can be set to identify easily the pass or fail.

All measurement results are logged for later detailed analysis. Remote controllability allows the user to operate the test remotely.

Clarity - Measuring The Human Perception

Clarity can also be described as speech intelligibility, indicating how much information can be extracted from a conversation. Speech intelligibility depends on a large variety of influencing factors such as quality of the speaker and microphone, speech codecs, compression, packetization in VoIP networks and the effects of packet-loss and jitter. Measuring just these effects is not sufficient because the human brain is able to compensate for some of these shortcomings. This requires a more sophisticated method of analysis.

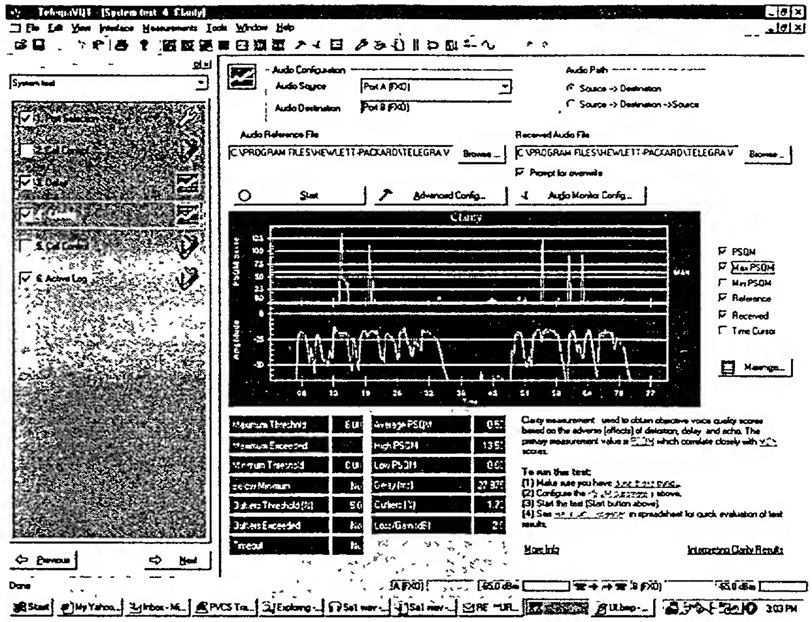
The Telegra VQT offers two innovative methods for measuring speech quality: PSQM+ and PAMS.

PSQM+

The Telegra VQT transmits actual human speech across the network and uses the industry standard ITU-T P.861 Perceptual Speech Quality Measure (PSQM) to objectively measure how clear the audio is at the receiving end. Designed for analysis ot compressed voice, PSQM is a cognitive model that objectively determines how people perceive the audio quality. Especially for the effects of voice over packet or cell networks the Telegra VQT uses an enhanced version of PSQM, known as PSQM+, in order to account for severe distortions and time clipping as experienced in packet networks.

The PSQM measurement is actually performed with real human speech by comparing the reference and received signals. The VQT offers 144 different voice samples in 8 different languages to allow you to measure the network behavior for many different users. The VQT also provides the equivalent Mean Opinion Score (MOS) for every PSQM measurement.

In addition to presenting the maximum, minimum and average PSQM score the Telegra VQT also provides a graphical representation of the PSQM scores over time during the entire speech sample, and reports the standard deviation. This allows the user to identify network effects that influence the speech quality such as packet loss. To compensate for network delay, the received signal is time aligned with the reference signal to allow an accurate PSQM measurement.



PAMS

The Perceptual Analysis Measurement System (PAMS) is a valuable tool for providing an objective measurement of speech quality. It uses a perceptual model based on human hearing factors, and provides a repeatable, objective means for measuring perceived speech quality. PAMS uses a different signal processing model than the ITU P.861 standard PSQM, and produces different types of scores. It provides a "Listening Quality Score" and a "Listening Effort Score", both which correlate to MOS scores on a 1-5 scale.

In addition to the listening scores, the VQT provides a graphical representation of signal loss and additive distortion over both the time and frequency domains of the test signal. This is known as the error surface. The error surface shows the impacts of a wide range of network-induced distortions, including coding distortion, front-end clipping, muting, noise, and bit or frame errors. The amplitude of errors is related to how audible and annoying they will be.

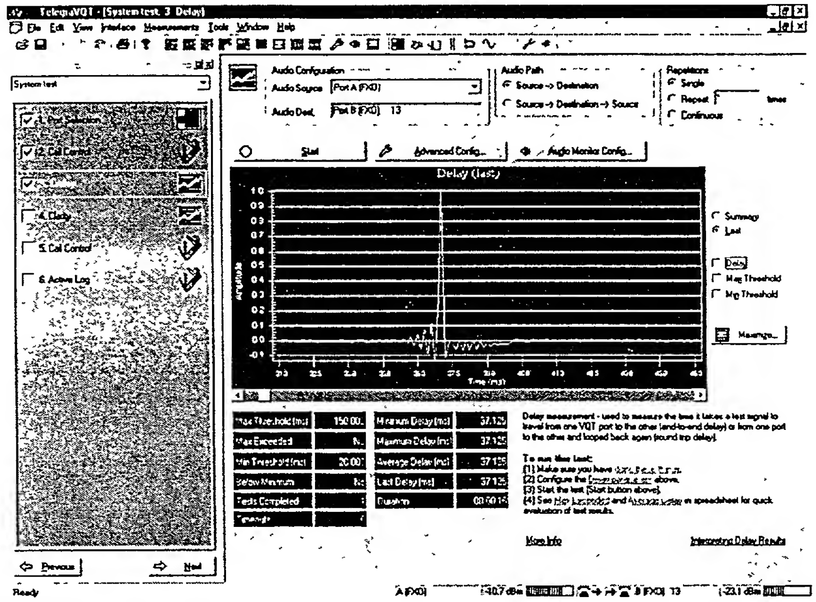
In addition to single clarity measurements Telegra VQT also allows trend analysis over a period of time. This provides essential data to understand network performance variations over the course of an hour, a day or more.

Delay - No Longer Just An International Long-Distance Problem

Delay is the time required for a signal to traverse the network. In a telephony context, end-to-end delay is the time required for a signal generated at the talker's mouth to reach the listener's ear.

Telegra VQT provides a very accurate way of measuring delay, via an impulse response measurement using a Maximum Length Sequence (MLS) noise burst. This pseudorandom noise appears like white noise, and allows the user to determine the delay behavior of a network across all frequencies. The impulse response is graphed and the user can visually inspect the delay results. The MLS signal enables highly accurate time-correlation of the transmitted and received signals, allowing the delay for the entire transmission of a signal to be accurately measured. Both end-to-end and round-trip delay measurements can be performed.

In addition to single delay measurement, Telegra VQT also allows to perform multiple delay measurements. It graphs delay over time and also calculates average, minimum and maximum delay.



Echo - Go Beyond Simple Detection

Echo is a phenonmenom introduced by hybrid wire junctions in circuit-switched networks. Echo can have a detrimental effect on voice quality if the delay and signal level are great enough. But until now, measuring that effect has been elusive.

The Telegra VQT provides two key echo measurements. The Perceived Annoyance Caused by Echo (PACE) measurement detects voice echo and determines the impact that echo has on a speaker's perception of voice quality. The Telegra VQT transmits a sample of human voice and measures the return echo. PSQM scoring is applied to the superimposition of the received echo on the originally transmitted voice, using the originally transmitted voice as the reference signal. The Telegra VQT presents useful information:

- The signal levels of both the transmitted signal and any received echo are graphed and presented in the time domain.
- PSQM scoring is graphed, correlated with the transmitted signal and the echo signal
- Average and maximum PSQM scores are individually reported
- Delay of tail-circuit echo is presented.
- Each duration of echo received during speech, and each duration of echo received during silence, are distinguished and graphed
- The total duration of echo received during speech, and the total duration of echo received during silence, are individually reported in milliseconds.
- The percentage of a voice transmission that is echo-free is reported.

Tail-end echo can be measured using the Telegra VQT's E&M ports, which ensure no origination-side echo will be encountered. Origination-side echo, or immediate echo, can be measured using the Telegra VQT's FXO ports. In addition, the Telegra VQT can simulate echo on its destination port. A network simulator function applied to the destination port can provide varying degrees of delay and return loss to a signal, to simulate echo and exercise the capabilities of echo cancellers.

The second key echo measurement is Echo Doubletalk. This measures the performance of echo cancelers in canceling the echo of one speaker's voice while passing through another simultaneous speaker's voice unimpaired. The VQT applies the condition of Doubletalk by transmitting simultaneous voice in both directions. It measures the clarity of the voice in one direction, the "Doubletalk" signal. Any impacts due to the uncanceled echo of the other speaker, or impairments on the Doubletalk signal, are detected and measured.

Simulate Real Network Behavior In The Lab

Testing systems under real-life network conditions in the lab can help to shorten deployment time and identify problems early. The approach is as follows:

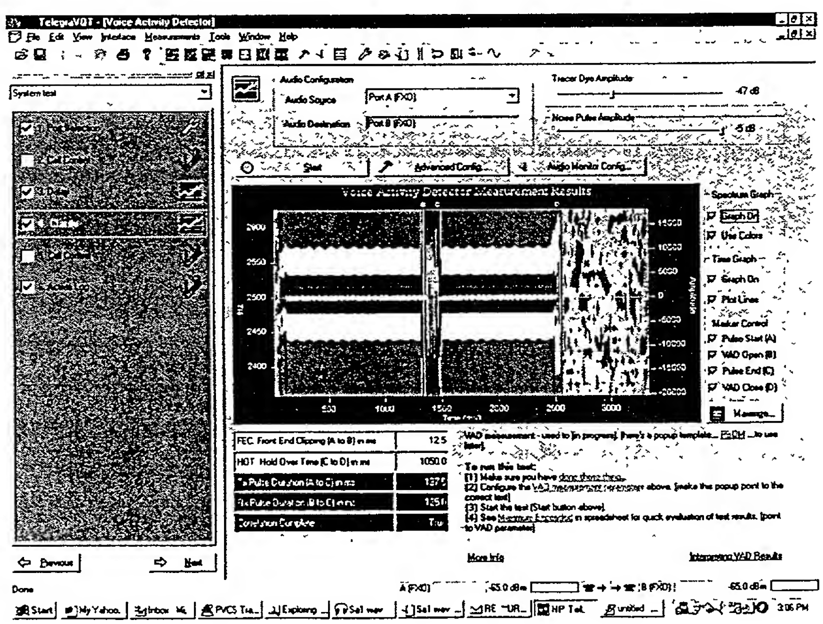
1. In the field, capture the network characteristic of an actual tail-end network by using the Telegra VQT impulse response measurement. Sending out white noise into the network, the transfer function of the network is captured. Telegra VQT presents the impulse response in the time domain measuring network delay.

2. Back in the lab the voice system such as a VoIP gateway can than be connected to the Telegra VQT analog ports. Using the previously captured network behavior Telegra VQT simulates the network behavior The impact on the voice quality can be determined and gain and delay can be varied to determine the influence of these factors.

This enables the developer or system integrator to determine the behavior of the voice system under real-life conditions right in the lab. The impact on the voice quality can be determined and changes to the design or system settings can be validated immediately. Changing gain or delay of the filter characteristics can simulate the effects of shorter or longer tail-end circuits, additional or less distance or number of nodes. It can also be used to determine boundary conditions under which the system is still working appropriately.

VAD

Voice activity detectors are implemented in voice gateways and are responsible for silence suppression and comfort noise generation. Silence suppression makes use of the fact that human conversations typically comprise more silence than speech from each speaker. Silence suppression stops digitizing when no voice signal is present. This can realize approximately 50% reduction in bandwidth requirements. During periods of this silence, the listener still expects to hear some background noise to confirm that the connection is still active. A Comfort Noise Generator at the receive-side generates a background noise signal matching the real noise on the line to the listener during these silence periods.



To identify the appropriate function of these network components the Telegra VQT provides a VAD measurement to accurately measure the behavior of the silence suppressor and the comfort noise generator. The VAD measurement determines the following parameters:

- Front-end clipping - the time the VAD needs to detect the speech signal and how much of this signal got cut off. Listeners can be very annoyed by Front-End-Clipping if it makes the first word of each sentence difficult to under stand.

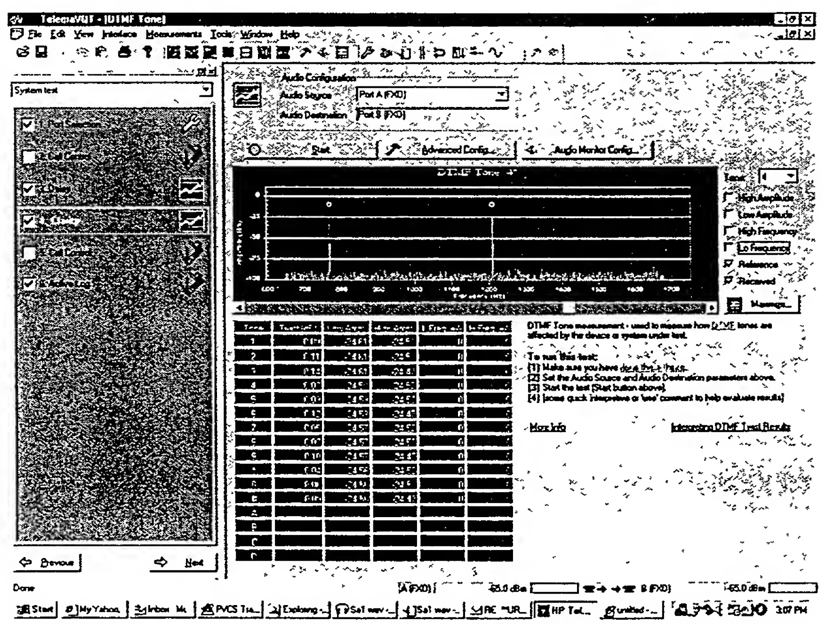
- Holdover time - the time the VAD continues to send data, even after the speaker stopped talking. A conservative/long holdover time will utilize bandwidth unnecessarily sending background noise; an aggressive/short holdover time will potentially stop the voice transmission even in short pauses between words. This can make the front-end-clipping very perceptible and annoy the listener.

- The match between the generated "comfort" noise and the "true" background noise. A noticeable difference in sound can be annoying to both speaker and listener

These VAD measurement results are presented graphically as well as in text form.

DTMF

DTMF stands for Dual-Tone Multi Frequency and is the tone generated by each key of a touch-tone phone. Transmitting DTMF tones through digital networks can be especially difficult with low bit-rate voice codecs, which are tuned to encode speech, a non-sine wave signal. DTMF, however, transmits two distinct sine-wave frequencies per key and low-bit-rate codecs often have difficulty recreating these signals. This can make it impossible to communicate with a voice message or interactive voice response system.



Telegra VQT provides the ability to determine how distorted these DTMF tone are when transmitted across a network. The system determines DTMF twist, the difference between the high frequency and low frequency amplitude. It presents send and received DTMF frequency to visualize the difference between the two tones. In addition it shows the results in tabular format, including amplitude and difference in peak frequency.

Automated Testing

The VQT's easy-to-use interface enables both interactive and automated testing The VQT's graphical task lists can be saved and executed automatically, with precise configurations set for each task list. This allows unattended testing and efficent high-volume testing. Execution can be via the GUI or a command line interface.

Related Literature

Telegra D	Product Overview	5968-5651E
Telegra M	Product Overview	5968-5652E
Telegra VQT	Technical Specification	5968-8811E

Warranty

Hardware: 1 year
Software: 90 day replacement only

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- ☐ Yes
- ☐ No
- ☐ In process

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- ☐ 180 days
- ☐ 90 days
- ☐ Other – (please define) ____________________

Product(s) of Interest

- ☐ ***The Agilent Advisor*** – Integrated, High-Performance Troubleshooting for:
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  - ____ Advisor ATM
- ☐ ***The LAN Analyzer*** – Scaleable Ethernet and Token Ring Test Solutions
- ☐ ***Telegra Fax Test*** – Fax Protocol and Low Generation Analysis
- ☐ ***Telegra Voice Quality Tester*** – Detailed Voice Analysis for Clarity, Echo and Delay using PSQM and PAMS
- ☐ ***Telegra Voice and Fax over IP*** – Protocol Analysis
- ☐ ***FASTest*** – Automated Service Verification for PSTN and IP Networks

What is the main problem you need to solve on your network?

____

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This Product is Y2K Compliant

Agilent Ordering Information

J1981A Telegra R Voice Quality Tester

- Opt. 200 Dual-port analog FXO and dual-port analog E&M interface
- Opt. 201 Dual-port T1 interface
- Opt. 202 Dual-port E1 interface
- Opt. 400 PAMS for clarity measurements
- Opt. 401 PSQM for clarity measurements

Instructor Led Training CBT

H7211B Essentials of VoIP Protocols

- Opt. 207 Instructor Led Training

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DATA SHEET

Cisco IP Telephone 7910 and 7910+SW

CISCO SECOND-GENERATION IP PHONES ONCE AGAIN ADVANCE STATE-OF-THE-ART TECHNOLOGY TO VOICE COMMUNICATION SOLUTIONS CISCO SYSTEMS, THE WORLDWIDE LEADER IN NETWORKING FOR THE INTERNET, NOW BRINGS TO MARKET NEW OPPORTUNITIES FOR RAPID DEPLOYMENT OF CLASSIC AND NEW WORLD VOICE APPLICATIONS BY PROVIDING HIGH-QUALITY VOICE INSTRUMENTS THAT USE IP TRANSPORT TECHNOLOGY. THIS ALLOWS THE CONSOLIDATION OF DATA AND VOICE INTO A SINGLE NETWORK INFRASTRUCTURE, INCLUDING A SINGLE CABLE PLANT; A SINGLE SWITCHED ETHERNET FABRIC FOR CAMPUS OR BRANCH OFFICES; AND UNIFIED SYSTEMS FOR OPERATIONS, ADMINISTRATION, AND MANAGEMENT (OAM) FOR DATA AND VOICE.

The basic feature member of the second-generation Cisco IP Phone portfolio is the 7910, primarily designed for common-use areas such as lobbies, break rooms, and hallways that require basic features This single-line phone also provides four dedicated feature buttons, located prominently under the display for Hold, Transfer, Call Park, and End Call. An additional group of feature access keys can be programmed by a system administrator. The standard configuration for these keys includes, *speed dial, redial, messages*, and *conference.*

The 7910 also provides a large character-based 2x24 character LCD display. The display provides features such as date and time, calling party name, calling party number, and digits dialed.

Additional buttons for call monitor speaker (used for on-hook dialing) and handset volume control, and a ringer and mute button for the handset microphone are arranged at the bottom of the set.

The Cisco IP Phone 7910 plugs into a standard RJ-45 Ethernet with one 10 BaseT interface. The 7910+SW model also supports 10/100 BaseT and has 2 RJ-45 connections.

The footstand of the 7910 is adjustable from flat to 60 degrees to provide optimum viewing of the display and comfortable use of all buttons and keys.

Basic Specifications:

- Hearing-aid-compatible (HAC) handset with ADA-compliant volume
- G 711 and G 729a audio compression
- H.323 and Microsoft NetMeeting compatibility
- Both DHCP and Boot P are supported
- Dynamic Host Configuration Protocol (DHCP) automatically assigns IP addressee to devices when you plug in the phone.
- Comfort noise generation and voice activity detection (VAD) programming on a system basis
- The 7910 is dynamic and designed to grow with system capabilities. Features will be able to keep pace with new changes via software updates from the system.

Physical Specifications

- Dimensions (H x W x D): 8* x 10 1/2 x 6 in. (20 32 x 26.67 x 15.24 cm)

*The footstand is adjustable from flat to a maximum angle of 60 degrees. In the flat position (for wall mounting), the phone measures 4.25 inches high. In the maximum upright position on a desk, the phone is 8 inches high.

- Phone weight· 2.2 lb (1.0 kg)
- Polycarbonate ABS plastic in textured dark gray

CISCO SYSTEMS

- One standard 10BaseT RJ-45 interface
- 48 VDC required, supplied locally at the desktop using an optional AC to DC power supply

Order

- CP-PWR-CORD-NA (North America)
- CP-PWR-CORD-CE (Central Europe)
- CP-PWR-CORD-UK (United Kingdom)
- CP-PWR-CORD-AU (Australia)
- CP-PWR-CORD-JP (Japan)
- CP-PWR-CORD-AP (Asia Pacific)

The Cisco IP Phone 7910 can also power down the LAN from any of the new inline power capable blades and boxes.

Temperature

- Operating temperature: 32 to 104 F (0 to 40 C)
- Relative humidity: 10% to 95% (noncondensing)
- Storage temperature: 14 to 140 F (–10 to 60 C)

Regulatory Compliance

- CE Marking

Safety

- UL-1950
- EN 60950
- CSA-C22.2 No. 950
- IEC 950
- AS/NZS 3260
- TS 001

EMC

- FCC (CFR 47) Part 15 Class B
- ICES-003 Class B
- EN55022 Class B
- CISPR22 Class B
- AS/NZS 3548 Class B
- VCCI Class B

Telecom

- FCC (CFR47) Part 68 (HAC)
- IC CS-03

Ordering Information Cisco IP Phone 7910 and Part Numbers:

- CP 7910 (includes Station User License)
- CP-7910 = (Spare phone, does not include Station User License).

For More Information on Cisco Products:

U.S. and Canada: 800 553-NETS (6387)
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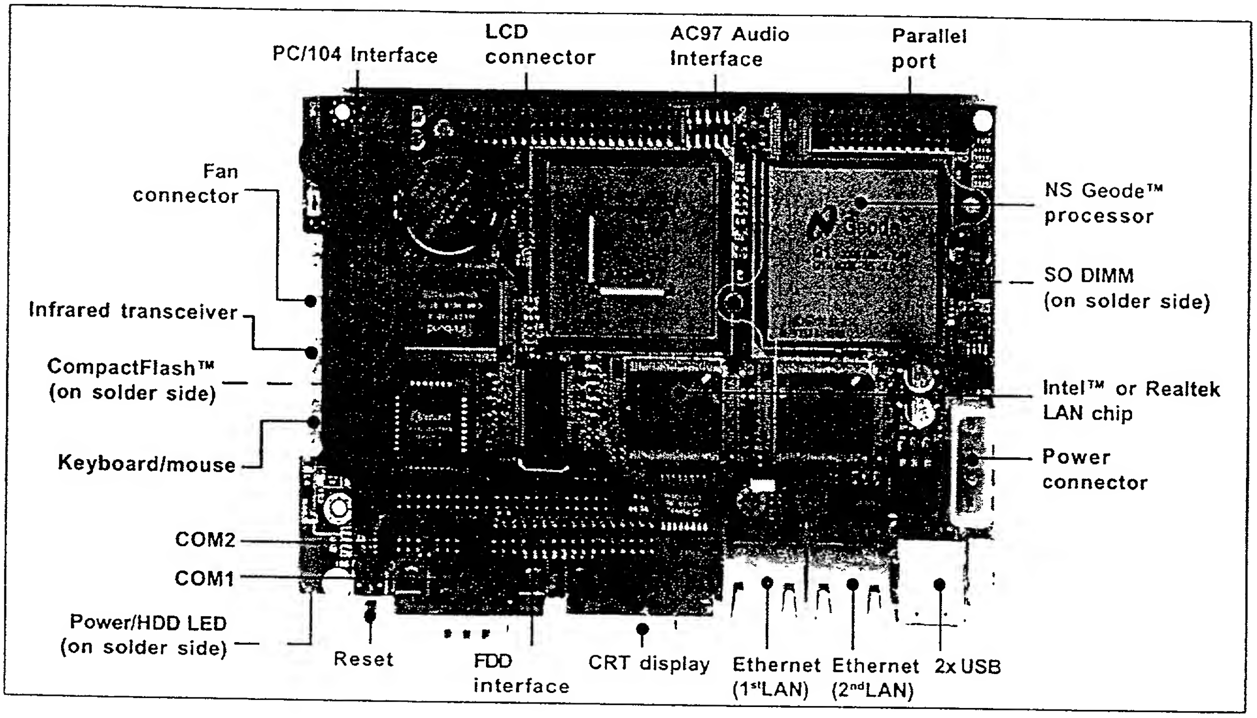
PCM-5823

NS Geode with Dual Ethernet, Audio, and VGA/LCD



Introduction

Advantech's new PCM-5823 NS Geode Series is a 3.5" SBC which offers an onboard NS Geode GX1-300 processor or GXLV-200 processor, plus support for dual ethernet for network functions such as Firewall, Router or Gateway functions. Other onboard features include VGA/LCD, a CompactFlash™ card socket, and watchdog timer. With the 586-level NS Geode processor mounted directly on board, upgrade and system configuration is much more convenient as is the benefits of fanless operation in temperatures up to 60° C (140° F). This board is a feature-packed, 586-level, hassle-free solution for space critical applications.

Dual-Ethernet Flexibility

The flexible, dual Ethernet capability of the PCM-5823 gives network administrators another tool to deal with today's changing application needs. The board comes with the top-of-the-line Intel 82559E LAN chips or cost-effective RTL 8139C LAN chips, depending on users needs. Used as a DMZ device, the PCM-5823 could act as a firewall that sits between the internet and a company's internal network. This applies to Web (HTTP) servers, FTP servers, SMTP (e-mail) servers or DNS servers. The PCM-5823 can also be used as an internet-network gateway (router) that connects a number of local networks. Once again, the dual ethernet capability and processing power of the PCM-5823 gives it the ability to perform these functions. The compact size, low power consumption and fanless operation also allow it to be used "exactly" where you need it as well.

Features

- Onboard NS GX1-300 MHz CPU or GXLV-200 MHz CPU
- Dual Ethernet On board
- Provides AC97 Audio interface (optional)
- Single +5 V power supply
- ISA-bus expansion with on-board PC/104 connector
- **Compact size:** 145 x 102 mm fits in the space of a 3½ " HDD

Specifications

- **CPU:**
  Onboard NS GX1-300 processor or Onboard GXLV-200 processor
- **BIOS:** AWARD 256 KB Flash memory
- **Chipset:** NS CX5530
- **System memory:** One 144-pin SO DIMM socket accepts up to 128 MB SDRAM
- **Enhanced IDE interface:** Supports up to two EIDE devices. BIOS auto-detect, PIO Mode 3 or Mode 4 transfer, Ultra DMA33 mode (ATA-4) up to 33 MB/sec
- **FDD interface:** Supports up to two FDDs
- **Serial ports:** One serial RS-232 port, one serial RS-232/422/485 port
- **Watchdog timer:** Software enable/disable intervals at 1.6 seconds

Specifications cont.

- **Parallel port**: One parallel port, supports SPP/EPP/ECP mode
- **Keyboard/mouse connector**. Mini-DIN connector supports standard PC/AT keyboard and PS/2 mouse
- **USB interface**. Two USB ports connectors on the front side
- **Infrared port**: Supports up to 115 Kbps transmitting rate
- **Power management**: APM 1.1 compliant power management
- **PC/104 expansion**: 104-pin 16-bit PC/104 module connector

Flat Panel/VGA Interface

- **Chipset**: NS CX5530
- **Display memory**: 1 ~ 4 MB shared with system memory
- **Display type**: Simultaneously supports both CRT and 18-bit TFT LCD displays (supports 3.3 V LCD)
- **Resolution**:
  - Non-interlaced CRT monitor resolution up to 1024 x 768 @ 16 bpp or 1280 x 1024 @ 8 bpp
  - Panel resolutions up to 1024 x 768 @ 18 bpp TFT panel

Solid State Disk

- Supports one 50-pin socket for CompactFlash card

Ethernet Interface

- **Chipset**: Intel 82559 for the PCM-5823-G0A1 only or the RealTek™ RTL 8139 chip for the PCM-5823-D0A1 only
- **Ethernet Interface**: PCI 10/100 Mbps Ethernet, IEEE 802.3U protocol compatible
- **Connection**: 2 RJ-45 connectors with LED on the frontside

Audio Function (optional)

- **Chipset**: NS CX5530
- **Interface**: Supports AC97 interface for optional audio module

Mechanical and environmental

- **Power requirements**:

  Maximum: 5 V @ 4 A (4.75 ~ 5.25 V)
  Typical: +5 V @ 1.5 A (for GX1-300 and GXLV-200)
- **Dimensions**: 145 mm x 102 mm (5.7" x 4")
- **Weight**: 0.77 kg (1.70 lbs) (weight of total package)
- **Power supply voltage**:
  +5 V (4.75 ~ 5.25 V)
- **Operating temperature**:
  0 ~ 60° C (32 ~ 140° F)

Ordering Information

- ❑ **PCM-5823-D0A1**
  NS Geode SBC with GXLV-200 CPU, VGA/LCD and dual RTL Ethernet
- ❑ **PCM-5823-G0A1**
  Same as PCM-5823 but with GX1-300 CPU and dual Intel Ethernet
- ❑ **PCD-100A-0004/0008/0016/0032/0048/0064/0096/0128**
  CompactFlash card with 4/8/16/32/48/64/96/128 MB memory capacity
- ❑ **9689000065**
  32 MB SDRAM SODIMM
- ❑ **9689000061**
  64 MB SDRAM SODIMM

Note: All features and specifications are subject to change without notice.
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